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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
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EXAMINER	
FLANDERS, ANDREW C	

ART UNIT	PAPER NUMBER
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	09/985,976	CORNELISSE, LEONARD E.
	Examiner	Art Unit
	Andrew C. Flanders	2615

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 20 July 2007.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-40,42,43 and 45-58* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-40,42,43 and 45-58* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08)
Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413)
Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 06 August 2007 has been entered.

Response to Arguments

Applicant's arguments with respect to the claims have been considered but are moot in view of the new ground(s) of rejection necessitated by Applicant's amendment.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claim 1 – 40, 42, 43 and 45 - 5~~7~~8 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ishige (U.S. Patent 5,892,836) in view of Melanson (U.S. Patent 6,104,822).

Regarding **Claims 1, 13 and 33**, Ishige discloses:

A method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (abstract), said method comprising the steps of:

(a) converting the acoustic input signal into a digital acoustic input signal (i.e. and input circuit for receiving the analog audio signal and converting it into a digital signal; Fig. 7 element 12);

(b) transforming the digital acoustic input signal into one or more frequency domain input signals (Fig. 6 and col. 7 lines 37 – 57);

(c) detecting the magnitude of each of the one or more frequency domain input signals (i.e. a frequency analyzer; Fig. 7 element 21).

Ishige does not explicitly disclose:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal.

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value.

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal:

(h) converting the digital acoustic output signal into the analog acoustic output signal.

Melanson discloses a digital signal processor hearing aid with a program selector switch (Fig. 1a element 46) that is preferably manipulable by a user to allow the user to dynamically select which of the digital signal processing means to invoke in which listening environment. In dealing with these environments, each of the processing means may implement such functions as *compression*, noise compensation, feedback cancellation, etc; col. 8 lines 30 – 50.

Applying this environmental selection switch to Ishige would allow the user to conveniently alter the characteristic's of Ishige's hearing aid to further assist the user in various environments.

Modifying Ishige to include the selection feature taught by Melanson discloses:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation (col. 8 lines 30 – 50; col. 19 lines 10 – 15 of Melanson; the switch of Melanson alters the characteristics such as

compression, since it is set for various environments, changing it will either directly increase it or decrease it accordingly to a set level);

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (i.e. the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10; this portion of Ishige functions as the various DSPs in Fig. 1a of Melanson);

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (i.e. the digital audio signal is supplied to the hearing compensating circuit; Fig. 7 element 22);

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal (i.e. the output of the hearing compensation circuit is applied to the output circuit; Fig. 7);

(h) converting the digital acoustic output signal into the analog acoustic output signal (Fig. 7 element 13).

It would have been obvious to one of ordinary skill at the time of the invention to apply the adjustable features of Melanson to the hearing aide of Ishige. Melanson discloses that previous hearing aids are typically capable of only providing one single strategy with adjustable parameters. This is similar to what Ishige discloses. Melanson further states that in the hearing aid of Fig. 1a, the hearing aid can implement more than

one strategy and thus is better able to adapt and to provide optimal results in a variety of different listening environments.

Regarding **Claims 2, 14, 34 – 37 and 57**, in addition to the elements stated above regarding claims 1, 13, and 33, the combination further discloses:

wherein step (d) further comprises adjusting the configurable input/output characteristic for at least one frequency band corresponding to the one or more frequency domain input signals by one of:

increasing the level of said configurable input/output characteristic by a larger amount for lower level sounds compared to higher level sounds when a user adjusts the user adjustable digital loudness normalization control signal to increase the level of the analog acoustic output signal,

and decreasing the level of said configurable input./output characteristic by a smaller amount for lower level sounds compared to higher level sounds when the user adjusts the user adjustable digital loudness normalization control signal to decrease the level of analog acoustic output signal (i.e. in the combination, Ishige's hearing aid is adjustable as taught by Melanson. Melanson discloses that the invention is adjustable to different listening environments thus optimizing the amplification to fit the environment; cols 8 and 9. Thus, adjusting the hearing aid of the combination increases and decreases the level of the input/output characteristic dependent upon the setting of the user in various environments).

Regarding **Claims 3, 15 and 50**, in addition to the elements stated above regarding claims 1, 13 and 33, the combination further discloses:

performing steps (c), (e) and (f) by means of a programmable processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals and are programmed via the memory through the fitting device).

Regarding **Claims 4, 21, 39 and 53**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises calculating the corresponding gain value for the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor, wherein a parameter of the fitting formula is provided by the user adjustable digital loudness normalization control signal (i.e. the hearing compensating filter coefficient setting circuit defines the coefficients for the filters in the hearing compensating circuit on the data that is in the memory which is supplied by the user (Fig. 7 and the associated text in the disclosure) through the input of Melanson)

Regarding **Claims 5, 22, 38 and 51**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a look-up table stored in said programmable digital signal processor, wherein information in the look-up table is

retrieved based on the user adjustable digital loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 6, 23, 24 and 52**, in addition to the elements stated above regarding claims 5, 22 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claims 7, 25, 40 and 54**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor and a look-up table, wherein information in the look-up table is retrieved based on the user adjustable digital

loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 8, 16, 17, 26, 27 and 55**, in addition to the elements stated above regarding claims 7, 5, 25 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claim 9**, in addition to the elements stated above regarding claim 1, the combination further discloses:

wherein step (b) comprises transforming the digital acoustic signal into at least two frequency domain input signals, each of said frequency domain input signals having a configurable channel input/output characteristic associated therewith, said configurable channel input/output characteristic together forming said configurable input/output characteristic, and wherein said at least two frequency domain input signals

are provided with different channel input/output characteristics (i.e. Fig. 6 and col. 7 lines 37 – 57; and the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10).

Regarding **Claims 10 – 12, 20, 28 – 30 and 45 - 49**, in addition to the elements stated above regarding claims 1, 13 and 34, the combination further discloses wherein said configurable input/output characteristic is a curvilinear compression characteristic, an input compression characteristic, and an output compression characteristic. Ishige discloses matching the input audio signal with the narrowed dynamic range of the person fitted with the hearing aid using a filter; col. 7 lines 4 – 7. Thus, depending on the users hearing loss characteristics, the device may increase or decrease the high and low frequency components at different values. Further portions will lower amplitude values may be increased or decreased accordingly. As such, Ishige anticipates this element of the claimed invention. Additionally, the device is adjustable as taught by Melanson above.

Regarding **Claims 18, 19, 31 and 32**, in addition to the elements stated above regarding claims 9, 2, 13 and 33, the combination further discloses:

wherein each of said configurable channel input/output characteristics are (is) varied in response to said user adjustable digital loudness normalization control signal (i.e. the memory is programmed/fitted to the hearing characteristic of the user; and the

hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7).

Regarding **Claim 56**, in addition to the elements stated above regarding claim 33, the combination further discloses:

(a) a microphone for receiving an input sound providing an analog input acoustic signal (Fig. 7 element 11)

(b) an A/D converter coupled to said sound energy signal and for reception device for receiving said analog input acoustic signal or an image of said analog input acoustic signal and coupled to said analysis filter for providing said digital acoustic input signal (Fig. 1 element 12)

(c) a D/A converter coupled to said synthesis filter for receiving said digital output acoustic signal and for providing an analog output acoustic signal (Fig. 1 element 13);

(d) a speaker coupled to said D/A converter for receiving said analog output acoustic signal and providing an output sound energy signal (Fig. 1 element 14).

Regarding **Claims 42 and 43**, claims 42 and 43 claim various methods of which to adjust the control signal, a variable resistor and a two-way switch which are not explicitly disclosed by the combination. However, Melanson discloses a user manipulable switch but does not provide details on its implementation.

Examiner takes official notice that various different switches exist including variable resistors (i.e. potentiometers as shown by Martin US 6,104,822 previously) and mechanical switches.

As Melanson is silent it is clear that the implementation of the switch is not a key feature. Many different switches exist and their implementations in the art of hearing aides (such as the one of the combination) do not produce any new or unexpected results. In other words, using a mechanical two-way or a potentiometer would not patentably distinguish the claimed invention from the prior art because it does not produce any new or unexpected result.

Thus implementing a programming means such as the method disclosed by Martin or a two way switch would have been obvious to one of ordinary skill in the art. One would have been motivated to do so to in order to effectively enter an input in to the combination's hearing aid.

Claim 58 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ishige (U.S. Patent 5,892,836) in view of Topholm (U.S. Patent 4,947,432).

Regarding **Claim 58**, Ishige discloses:

A method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (abstract), said method comprising the steps of:

(a) converting the acoustic input signal into a digital acoustic input signal (i.e. and input circuit for receiving the analog audio signal and converting it into a digital signal; Fig. 7 element 12);

(b) transforming the digital acoustic input signal into one or more frequency domain input signals (Fig. 6 and col. 7 lines 37 – 57);

(c) detecting the magnitude of each of the one or more frequency domain input signals (i.e. a frequency analyzer; Fig. 7 element 21).

Ishige does not explicitly disclose:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to provide the user with a continually variable control for increasing or decreasing the amount of compression during operation.

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal.

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value.

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal:

(h) converting the digital acoustic output signal into the analog acoustic output signal.

Topholm discloses a programmable hearing aid with a multifunction sliding switch (Figs. 3 and 5).

Applying these sliding switches to Ishige would allow the user to conveniently alter the characteristic's of Ishige's hearing aid to further assist the user in various environments.

Modifying Ishige to include the selection feature taught by Topholm discloses:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for controlling the configuration of said input/output characteristic having an amount of compression for loudness normalization, the control signal being configured to be directly increased or decreased by the user for increasing or decreasing the amount of compression during operation (Sliders 12 of Topholm added to Ishige which allow for adjustment of such elements as compression, frequency and output power; col.3);

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (i.e. the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10 in Ishige);

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (i.e. the digital

audio signal is supplied to the hearing compensating circuit; Fig. 7 element 22 in Ishige);

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal (i.e. the output of the hearing compensation circuit is applied to the output circuit; Fig. 7 in Ishige);

(h) converting the digital acoustic output signal into the analog acoustic output signal (Fig. 7 element 13 in Ishige).

It would have been obvious to one of ordinary skill at the time of the invention to apply the adjustable features of Topholm to the hearing aide of Ishige. One would be motivated to do so to apply a known technique (Topholms adjustable characteristics) to a known device (Ishige's hearing aid and Hearing aids in general) ready for improvement (i.e. user adjustment) to yield predictable results (i.e. to allow user adjustment of Ishige's hearing aid as is notoriously well known desirable feature in the art).

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure: Brillhart (U.S. Patent 5,303,306).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Andrew C. Flanders whose telephone number is (571) 272-7516. The examiner can normally be reached on M-F 8:30 - 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Sinh Tran can be reached on (571) 272-7546. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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